

VOICE OVER INTERNET PROTOCOL

RELATED APPLICATION

This application is related to co-pending application Serial No. 09/666,758, filed on 9/21/00.

BACKGROUND

Systems have been designed to allow telephone subscribers to place calls directly, without the assistance of an attendant, after calling the main telephone number of a business. With the growing popularity of low cost DTMF (dual-tone multi-frequency) dial systems in the late 1970s and early 1980s, various solutions have been proposed to encode the dialed digits and then translate them to subscriber's names and refer callers to a database for a list of numbers served by the PBX (private branch exchange) of the called party.

The United States patent to Carter No. 4,608,460 is directed to such a system. Because a DTMF dial, however, is limited to twelve keys, a variety of translation schemes have been employed to make the translation. The end result, essentially, is always the same. The dialed digits represent an approximation of the alphabetically stored data for the intended party's name; and often this requires several attempts before the desired party can be located. If the directory of potential called parties is relatively limited, the system of Carter can be effective, and employs vocally conveyed information from a speech synthesizer

1 relating the other subscriber to the caller; whereupon the caller
2 selects the other subscriber (called party) in response to the
3 vocally conveyed information to effect the connection.

4 In the United States patent to Bourg No. 4,734,931, a
5 distributed calling directory uses a computer interface to a
6 central database to present the information to the calling party's
7 computer terminal to assist with the dialing. This system is
8 particularly useful by dedicated users, such as hotel desk
9 managers, to look up information in order to intelligently process
10 calls. The system, however, does not automatically provide inbound
11 callers with the directory information.

12 An effort to resolve the problems associated with DTMF dialing
13 is disclosed in the patent to Padden No. 4,979,206. In the system
14 of the Padden patent, a caller requesting directory assistance is
15 connected to an automatic speech recognition unit, and is prompted
16 to speak commands for identifying the requested directory number.
17 A speech recognition unit in the system converts the received
18 speech signals into data signals for searching a directory number
19 database. If a directory listing is located, the number of that
20 listing is announced to the calling party; and the calling party is
21 prompted to speak a command indicating whether a call to that
22 number should be established. Because of the wide variety of names
23 and pronunciation of names which exist, the system of the Padden
24 patent, and systems similar to it, are subject to shortcomings in
25 accuracy, and often send callers to an attendant or operator
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1 because the required name did not appear on file through the spoken
2 name.

3 A different approach is disclosed in the United States patent
4 to Rondeau No. 5,850,433. In the system of this patent, the video
5 display of a personal computer is employed to provide an
6 immediately visible directory reference from a computer server
7 database. The computer user is then allowed to select the desired
8 directory entry and to place a call to the intended party using a
9 voice link over a second communications line, or an encoded voice
10 stream directly to the personal computer. The approach used in the
11 system of this patent, however, requires a separate computer
12 database which often contains additional clutter, such as
13 advertising, with no integration with the main telephone or
14 employee directory of a business or other enterprise.

15 The United States patent to Brown No. 6,014,711 is directed to
16 a voice over Internet identifier for a directory to allow voice
17 messages to be sent to a recipient over an Internet mail connection
18 when at least one unique identifier for the recipient is known, but
19 when the electronic mail address of the recipient is unknown. The
20 identifier used may be the telephone number of the recipient, which
21 then is used by the system to translate the link to the e-mail
22 address. The user then may complete the link by addressing the
23 recipient at the e-mail address. Basically, the system of this
24 patent is an information message to the calling party desiring to
25 effect a voice connection over an e-mail address link.

26 Another approach is disclosed in the United States patent to

1 Nishimura No. 5,999,609. This patent is directed to a system for
2 processing call requests through the telephone network, under the
3 direction of the caller, who graphically manipulates the call
4 through a graphical user interface on a personal computer. The
5 system requires a personal computer display.

6 The United States patent to Furman No. 6,049,594 is directed
7 to a system for automatic vocabulary generation for voice dialing.
8 The system of this patent utilizes the storage of the record of the
9 past calling records of each calling party to determine the called
10 parties most likely to be called by this calling party. The voice
11 recognition system then operates, first with the previously stored
12 called parties to determine whether there is a match for any new
13 call.

14 It is desirable to provide an on-line directory service for a
15 business or enterprise which allows callers using either PCM
16 telephones or IP telephones or computers to call others within the
17 business or enterprise through access to an enterprise-wide
18 database of directory information, including translation tables for
19 communicating between PCM and IP devices.

20 SUMMARY OF THE INVENTION

21 It is an object of this invention to provide an improved
22 enterprise or business directory system and method.

23 It is another object of this invention to provide an improved
24 voice over Internet protocol (VOIP) directory which, upon
25 acceptance from the caller to place a call, forms automatic
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1 translation to an appropriate format to place a call attempt in
2 real time.

3 It is an additional object of this invention to provide a
4 voice over Internet protocol (VOIP) gateway integrated with the PBX
5 of an enterprise to allow callers from either PCM telephones or
6 Internet protocol (IP) telephones to locate and attempt live
7 contact with a person within an enterprise who has been listed
8 within the translation tables of a directory in the VOIP gateway.

9 In accordance with a preferred embodiment of the invention, a
10 voice over Internet protocol (VOIP) system for supporting real time
11 communications for both pulse code modulation (PCM) telephone
12 networks and Internet protocol (IP) telephone packet networks is
13 provided. The system includes a directory of PCM, IP and URL
14 addresses. IP addresses could include URLs as is used in SIP. It
15 also includes a translation address database for members of the
16 enterprise having IP terminals, addresses for IP telephones with
17 Internet connectivity, and addresses of other gateways on the
18 Internet. A VOIP gateway is linked with the directory, and is
19 connected to receive requests from both PCM and IP telephones. The
20 VOIP gateway provides voice prompts and responses to calling party
21 requests in the form of either DTMF digits or voice signals. The
22 VOIP gateway also operates in response to a calling party approval
23 to automatically process the link between the calling party and the
24 called party using, when necessary, translated addresses assigned
25 in the directory database, to the called party.
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1 BRIEF DESCRIPTION OF THE DRAWINGS

2 Figure 1 is a schematic diagram of a preferred embodiment of
3 the invention;

4 Figure 2 is a schematic diagram of another preferred
5 embodiment of the invention; and

6 Figure 3 is a flow chart describing the method of operation of
7 the embodiments of Figures 1 and 2.

8 DETAILED DESCRIPTION

9 Reference now should be made to the drawings, in which the
10 same reference numbers are used throughout the different figures to
11 designate the same or similar components. Figure 1 is a block
12 system diagram of a preferred embodiment of the invention in a
13 configuration best suited for a business or enterprise with a
14 single location and no web or Internet client access for real time
15 calling. The system shown in Figure 1 includes several different
16 parts which already exist in conjunction with a small business
17 application, such as a PBX system 10 coupled with a public switched
18 telephone network (PSTN) 12. In addition, the business includes a
19 local area network (LAN) 36 coupled through a firewall 42 to the
20 Internet or worldwide computer network 44.

21 The PSTN 12 is connected, as is well known, to public
22 telephones, such as public telephone 14 and public telephone 16,
23 representative of all of the various public telephones which have
24 specific addresses or numbers for communicating with one another
25 through the PSTN 12, and through the PSTN 12 to telephones for the
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1 business or enterprise served by the PBX system 10. The PBX system
2 10 of the small business or enterprise is shown as coupled with
3 work stations and telephone keysets 18 and 20, which are
4 representative of the various connections handled through the PBX
5 system 10 at that business or enterprise. In addition, facsimile
6 receivers 22 are served by the PBX system 10. Typically such
7 systems also include unified messaging units 24 operating with a
8 unified messaging database 26 in a conventional manner to provide
9 messages for stations which do not answer calling party requests.

10 In the system shown in Figure 1, the PBX 10 also is coupled
11 with a voice over Internet protocol (VOIP) gateway 22 through a T-1
12 trunk 30 and a CTI trunk 28. The VOIP gateway 32 is connected to
13 answer both calls from the PSTN 12 and calls that are transferred
14 from the local PBX 10, as indicated by the various interconnections
15 in Figure 1. The locally installed directory in the gateway 32
16 contains a translation database for members of the business
17 enterprise coupled with a local area network (LAN) 36, which has IP
18 terminals, addresses for IP telephones (display keyset devices on
19 the LAN) with Internet connectivity, and the addresses of other
20 VOIP gateways on the Internet, such as the gateway 50 shown coupled
21 to the Internet 44 in Figure 1. Also coupled to the LAN 36 are a
22 web page server 38 and an application server 40 which operate in a
23 conventional manner for the business enterprise.

24 The directory service, coupled with a local IP translation
25 database 34 in the VOIP gateway 32, allows a caller connected with
26 the VOIP gateway 32, from any one of the variety of sources already

Once the keyed in code or voice input has been received by the VOIP gateway 32, the speech decoder in the gateway or the DTMF decoder responds with the nearest match to the selection which is found in the directory database in the VOIP gateway 32. The response is a voice response, from either stored speech provided by the user or a text-to-speech representation of characters entered into the database. In addition, the system employs both "find next" and "find previous" functions to accurately access the desired name through communication with the calling party. Once the desired name has been heard from the built-in voice response unit (VRU) in the VOIP gateway 32, the caller gives approval in a conventional manner. The call then is processed in real time using the translated address in the directory assigned to the selected VOIP device. Disconnection is provided at the end of the call, either by encountering a disconnect message or loss of the data from the VOIP device, or by receipt of a disconnect signal from the PSTN/PBX connections.

It should be noted that the system allows seamless communication between devices of different types through the VOIP gateway 32. In the system shown in Figure 1, the VOIP gateway unit

1 32 combines the functions of gateway VRU and directory lookup
2 tables, rather than relying on physically separate servers for the
3 directory and the VRU. In the embodiment shown in Figure 2, the
4 system may be expanded to a multi-machine architecture when
5 installed as part of a larger business enterprise.

6 The VOIP gateway unit 32 of Figure 1 includes a control
7 chassis with a CPU, memory, mass storage devices, disk drives,
8 digital signal processor (DSP), arrays to convert between voice and
9 data signals, an operating system, and various software
10 applications to operate the gateway 32 in accordance with the
11 installed application. These components are conventional; and
12 their implementation in the system shown in Figure 1 is effected in
13 order to achieve the real time communication between conventional
14 pulse code modulation (PCM) telephones and Internet protocol (IP)
15 telephones operating over packet networks.

16 Storage of the user directory is provided within the control
17 chassis of the VOIP gateway 32, and is logically separated into one
18 database record per user. This means that each station has a
19 single database record in the directory, whether that station is
20 one connected to the PBX system, such as the stations 18 and 20, or
21 is a station coupled with the LAN 36 of the business enterprise.
22 Each record in the directory contains the appropriate fields to
23 uniquely identify the user by name and number codes required to
24 translate between the PSTN and LAN environments. The directory is
25 integrated with the user's computer network in such a way that it
26 can be manually administered or automatically updated, as the PBX

1 unified messaging and network addresses change their assignments.
2 The manner of updating the directory is not important to an
3 understanding of this invention, since directory updates can be
4 effected in a variety of well known conventional ways.

5 A typical operation of the system involves the translation of
6 calls coming into the business enterprise LAN-based terminals from
7 the PSTN 12 and the placement of calls from the LAN 36 out to the
8 PSTN. The VOIP gateway 32 performs the function of translation
9 from telephone number to IP address, and vice-versa, in a manner
10 which is transparent to the users of the system.

11 The system shown in Figure 1 provides an established path
12 between employees who are working at home, or are connected to the
13 Internet 44 while traveling. For example, such employees may have
14 an employee work station 48 or an IP keyset 46 coupled to the
15 Internet 44, or may be located at any location where they may
16 connect with the Internet 44 in any conventional manner. Because
17 such devices use the Internet protocol (IP), they require the
18 services of the gateway 32 and its included directory to place
19 calls to either the PSTN 12 or stations connected to the PBX 10.
20 For example for a request from the IP keyset 46, the VOIP gateway
21 32 plays the voice prompts over the IP stream, through the LAN 36
22 and the router firewall 42, through the Internet connection before
23 making a conversion to PCM for calls to standard telephone devices,
24 such as located on the PSTN 12 in the form of the devices 14 or 16,
25 or on the PBX system 10, such as the devices 18 and 20. However,
26 if an Internet caller selects a directory entry for another IP

1 device on the LAN 36, or coupled through the Internet 44, such as
2 the work station 48, the address simply is passed to the remote
3 party and the call is placed with no speech conversion to the
4 selected terminal.

5 The system shown in Figure 1 works particularly well for
6 business enterprises located all in one building connected to the
7 PBX (and, of course, coupled as described above through the
8 Internet 44, such as the stations 46 and 48), and who do not need
9 to provide real time calling services for web or Internet clients.
10 However, as business needs grow, the customer may require
11 conversion to a larger configuration of the directory service.

12 It also should be noted that optimum efficiency of the system
13 at the VOIP gateway 32 is achieved by using a variety of
14 conventional directory lookup techniques combined with efficient
15 data compression and a priority scheme. Once again, such data
16 compression and priority schemes are well known; and the details
17 are not necessary to an understanding of the invention. Basically,
18 the system of Figure 1 is designed to support twenty or so
19 simultaneous conversions and an equal number of simultaneous
20 directory lookup or connection requests.

21 For larger business enterprises having multiple locations and
22 many web client or Internet client access points for real time
23 calling, a larger network configuration is more suitable than the
24 system of Figure 1. Such a larger network configuration is shown
25 in Figure 2. The components of Figure 2 which are the same or
26 similar to those of Figure 1 are provided with the same reference

1 numbers. Like the system of Figure 1, the expanded configuration
2 of the system in Figure 2 serves as a directory service to directly
3 connect a variety of callers to different users within the business
4 enterprise. The business enterprise of Figure 2 is associated with
5 the PBX system 10. However, the directory server and the VRU have
6 been separated from the VOIP gateway 32 and are shown as a
7 directory server 84 coupled with a system directory data memory 86
8 and a VRU 82, coupled with the LAN 36 as shared resources. To
9 system users of either the system of Figure 1 or the system of
10 Figure 2, the operational characteristics are not changed; but the
11 quantity of stations or people served and the number of enterprise
12 locations have been increased in the system shown in Figure 2. As
13 shown in the system of Figure 2, the VOIP gateway 32 also is
14 responsive to public directory users, such as 54 and 56, operating
15 through the Internet 44 and the firewall 42 to the LAN 36 to access
16 the directory of the business enterprise served by the system.

17 In addition, the system of Figure 2 illustrates links to other
18 locations of the business enterprise through a corporate, private,
19 wide area network (WAN) 60 for different office locations, such as
20 62 and 72, which allow use of the system by private directory users
21 such as 64 and 74 at those different office locations, as well as
22 by IP keysets, such as the keyset 66, which are similar to the
23 keyset 46 described previously operating through the Internet 44.

24 Whether the call requests are made from or to the corporate
25 WAN 60 or through the Internet 44 or the LAN 36, to or from the
26 PSTN 12 or the PBX system 10, the operation of the VOIP gateway 32

1 is the same as described previously in conjunction with the system
2 of Figure 1. In the system of Figure 2, the directory database
3 also may be expanded. It is designed to look up data on the main
4 computer system of the business enterprise, as well as the main PBX
5 and unified messaging systems described in conjunction with the
6 system of Figure 1. These database links, coupled with the basic
7 directory structure for users are accomplished by relational links
8 to designated key fields within the data structure. This allows
9 the system to be administered using methods which are consistent
10 with back-end database maintenance methodology employed within most
11 business organizations. The system directory data in the directory
12 memory 86 is diagrammatically illustrated in the form 88 in Figure
13 2, and is linked with the unified messaging database 26, and with
14 the business enterprise data (ERP data) 92 administered by the
15 database server 90 coupled to the LAN for the business enterprise.

16 Reference now should be made to the flow chart of Figure 3,
17 which illustrates the various steps which take place in accordance
18 with call flow process, whether a call is originating from an IP
19 device, from the PSTN 12, or from a station on the PBX system 10.
20 All of the on-line translations occur in accordance with the steps
21 illustrated in Figure 3. Calls are either initiated by an IP
22 device 100 or a PSTN/PBX station 102. However a call is initiated,
23 it is directed to the gateway, which plays the directory prompt at
24 104 for ascertaining the identification of the station or called
25 party desired. The calling party then provides either DTMF digits
26 or voice signals at 106, and the gateway 32, in response to these

1 signals, searches the directory to determine whether there is a
2 match at 108. A voice response is provided at 108 for the closest
3 match; and the calling party negotiates an acceptable name at 110.
4 This is accomplished in a standard manner, and includes prompts for
5 both "find next" and "find previous" functions, as required, to
6 determine the acceptable name.

7 It should be noted that at this point, the system also may
8 operate in a conventional manner to divert the call to an
9 intervening operator if no acceptable match can be found. Since
10 such operator diversion is well known, it has not been indicated or
11 illustrated in Figure 3.

12 Assume that an acceptable name is found at 110, the calling
13 party (either at 100 or 102), then indicates selection of the name
14 at 112. The VOIP gateway 32 looks up the address translation in
15 the directory, where such a translation may be required, and
16 automatically translates the address to the appropriate phone
17 number at 116 and attempts the call placement at 118. This is a
18 real time operation, and functions to seamlessly interconnect IP
19 devices and PCM devices in a manner such that neither party is
20 aware that this is not a PCM-to-PCM or IP-to-IP link-up. As stated
21 previously, the VOIP gateway 32 includes a digital signal processor
22 (DSP), and arrays to convert between voice and data signals in a
23 real time on-line basis. The system for performing this conversion
24 may be of the type, for example, disclosed in the co-pending
25 application mentioned above.

26 The system which is disclosed in both Figures 1 and 2 also

1 enables the directory managed by the VOIP gateway to be accessed,
2 at least in part, by users browsing the directory on the Internet
3 44 and linking with the information on the web page server 38, as
4 well as callers who are processing a dialog with the VRU function
5 described previously. The data structure employed contains
6 appropriate fields to integrate both functions by using a single
7 administration point, namely the VOIP gateway 32.

8 It should be noted that the system also may be structured
9 using alternate implementations of the IDS database, including
10 those specified by standards like low-level directory access
11 protocol (LDAP). Also, other database implementations, such as the
12 Microsoft® Active Directory within Windows 2000® may become the
13 defacto standard for a single point of directory control within a
14 business enterprise, and may be used by both embodiments of this
15 invention. It also is envisioned to use a database function
16 provided by an Internet URL address to create a query to an
17 external database service. Because of these possible different
18 implementations, the actual implementation of the directory is not
19 confined to a dedicated directory server, such as the server 84
20 shown in the embodiment of Figure 2. A business enterprise
21 directory repository could well reside within the definition of a
22 multi-node computer operating system network with custom fields
23 added to the database scheme. It also is possible to embed the VOIP
24 gateway in the PBX or integrate it with a computer setup. Such
25 implementations are considered to come within the scope of this
26 invention.

1 The foregoing description of the preferred embodiments of the
2 invention is to be considered as illustrative and not as limiting.
3 Various changes and modifications will occur to those skilled in
4 the art for performing substantially the same function, in
5 substantially the same way, to achieve substantially the same
6 result without departing from the true scope of the invention as
7 defined in the appended claims.
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